NORTHERN METROPOLITAN COLLEGE OF TAFE

Collingwood Campus

## VIDEO EQUIPMENT TECHNIQUES T1

# AUDIO

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# AUDIO

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## Microphone Types.

A microphone is an instrument to produce electrical impulses corresponding to the frequency of the sound waves falling on it.

The three main types of microphones used in the television are:-

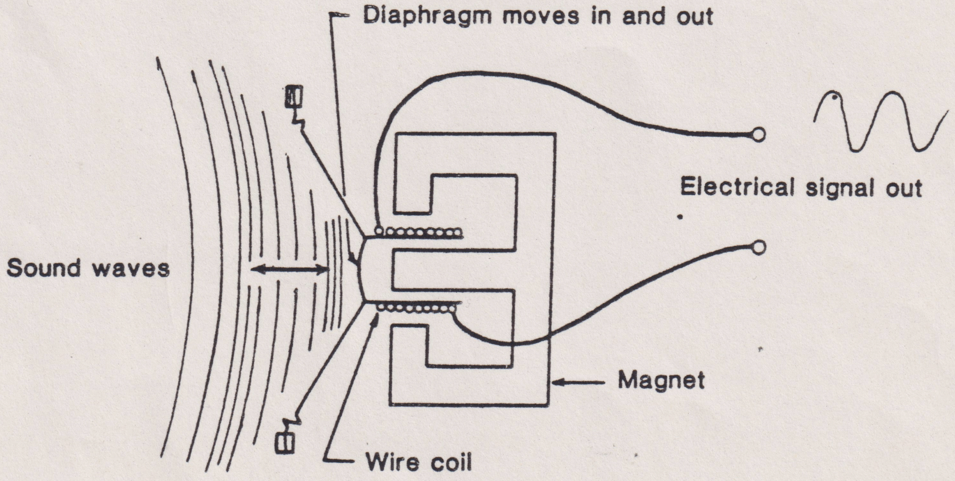
- (a) Dynamic.
  - (i) Coil.
  - (ii) Ribbon.
- (b) Condenser.
- (c) Electret.

### (a) Dynamic Microphone.

#### (i) Coil Type.

This type of microphone has a coil attached to a diaphragm (figure 1). When the sound pressure waves reach the diaphragm the pressure will cause the diaphragm to be pushed in, when the rarefaction part of the sound wave reaches the diaphragm then the tension will be released and the diaphragm will move outwards. Therefore the diaphragm is moved in and out according to the sound wave reach it.

The coil which is attached to the diaphragm slides over a central pole of a permanent magnet which induces a voltage into the coil which is an electrical replica of the sound wave.

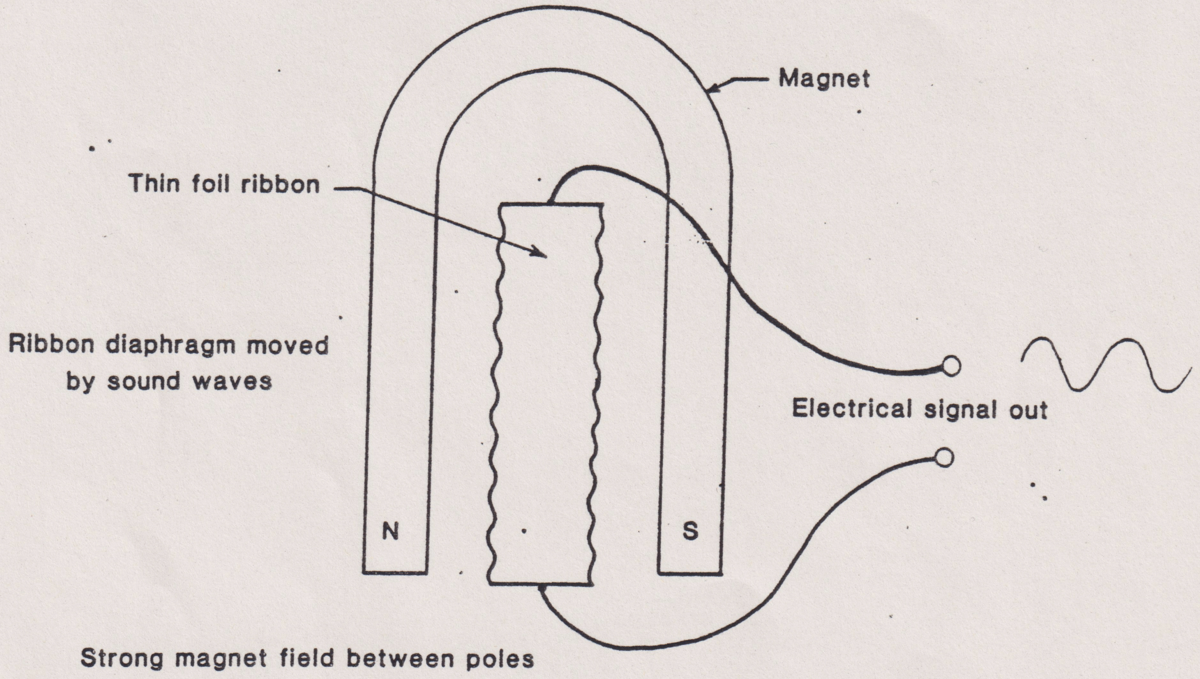


This type of microphone is used with most portable sound equipment because it requires no battery, extremely robust and are less sensitive to handling noise.

(a) Dynamic Microphone. (Continued)

(ii) Ribbon Type.

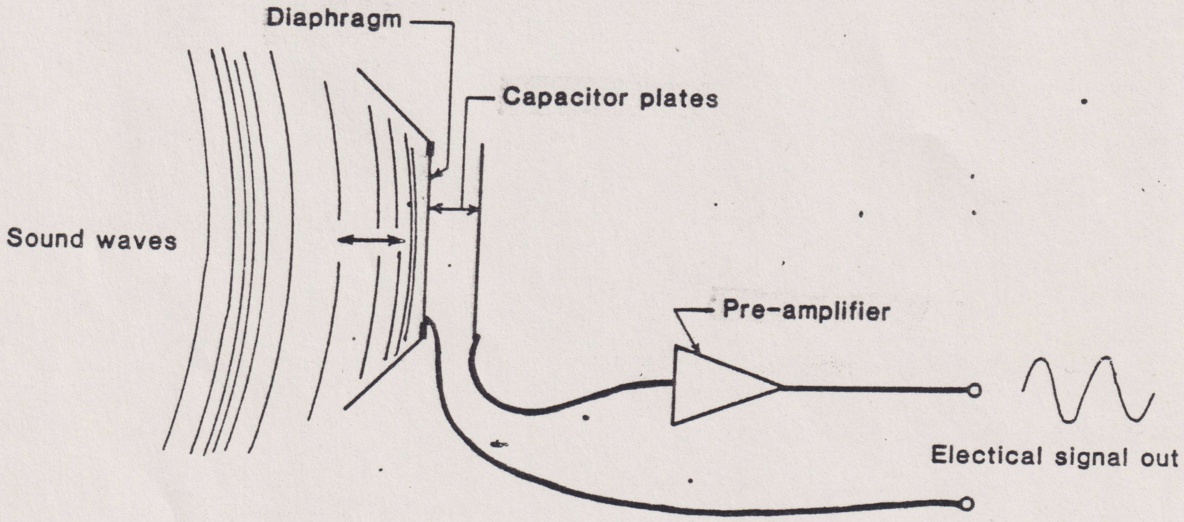
The ribbon type microphone (figure 2) consists of a corrugated light alloy foil diaphragm to which the electrical connections are made. The diaphragm is placed in a magnetic field from a permanent magnet, and an output voltage is generated as the ribbon diaphragm vibrates (due to sound pressure waves) in the field.


Magnet
Thin foil ribbon
Ribbon diaphragm moved by sound waves
Electrical signal out
N
S
Strong magnet field between poles

This type of microphone normally used audio booth.

## (b) Condenser Microphones.

In this type of microphone the diaphragm is used as one plate of capacitor and the back plate as the other (figure 3). A polarised voltage is applied to the plates. When the sound pressure waves move the diaphragm it causes the capacitance to vary, thus causing a small varying voltage which is then fed into a small amplifier built into the handle of the microphone which amplifies the small signal to a suitable level to be transmitted through the cable to the recorder.



Because of the power requirements (battery or phanton power supplies) and the higher cost they are being replaced by the dynamic microphone despite their superior higher sound level capabilities without distortion, superior high frequency and transient response characteristics.

## (c) Electret Condenser Microphone.

The difference between this type of microphone to the "normal" condenser microphone is that the diaphragm consists of a metalized plastic foil with a permanent electrostatic charge which allows the microphone to be minaturized and operate on low voltage.

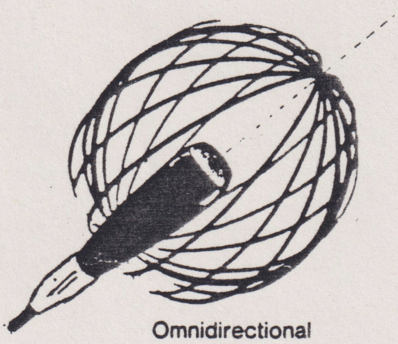
They are used in most forms of microphones from the tie clip type through to shotgun types. Their main disadvantage is they are battery powered and therefore require regular battery checks.

## Microphone Directional Characteristics.

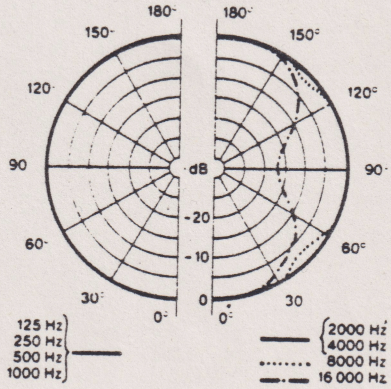
The directional response or field pattern of a microphone can be demonstrated by means of a polar diagram (figure 4). The polar diagram shows the sound pick-up sensitivity plotted against the angle from the main axis at a number of frequencies.

The main categories of directional characteristics are as follows:-

(i) Omnidirectional - receives sound uniformly from all directions.



Omnidirectional



Usually is somewhat limited in the distance to which the sound can be expected to give a reasonable output, approximately 30 cm. This type generally gives a smoother frequency response and is less susceptible to "popping" (i.e. letter p). Because the omnidirectional microphone picks up sound from all directions, background noise produced within its pick-up range can cause problems.

(ii) Bi-Directional. (Figure of Eight Response) - picks up the sound from front and back but not from the sides.

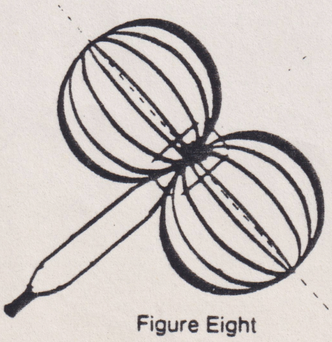
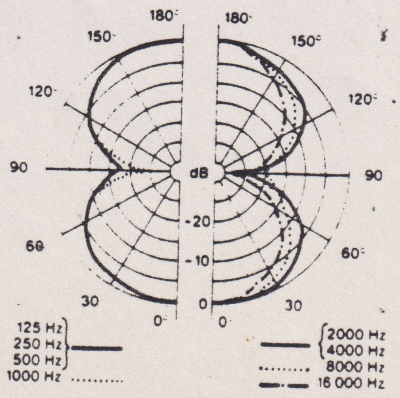


Figure Eight



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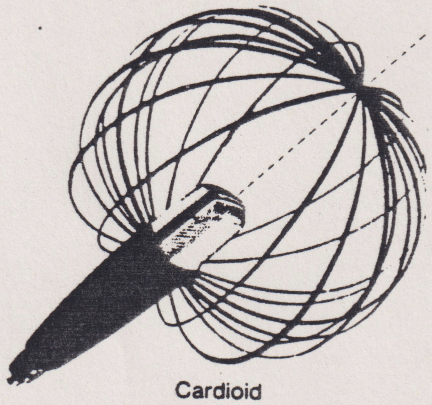
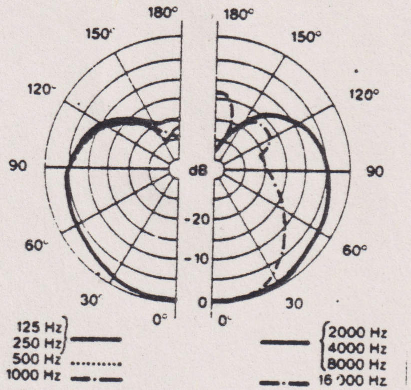
## Microphone Directional Characteristics. (Continued)

### (iii) Directional Microphones. (Unidirectional)

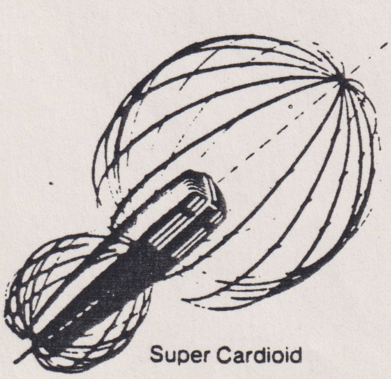
(a) Cardioid response pick-up pattern is a heart shaped (hence its name). Sound originating from the front of the microphone is emphasized.



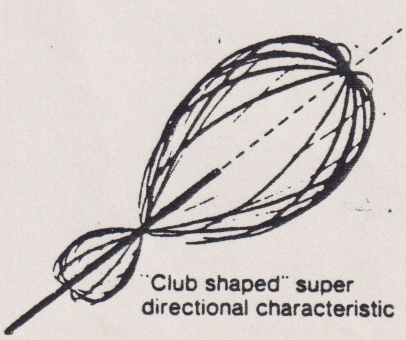
Cardioid



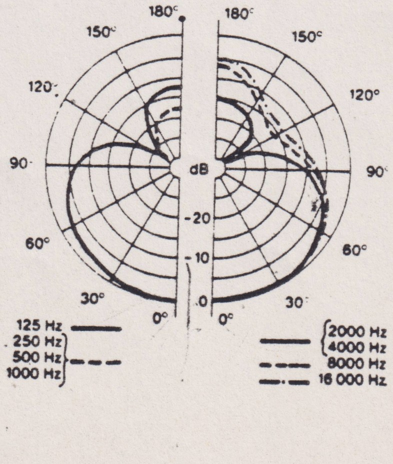
(b) Supercardioid response has a heart shape pick-up pattern as above, but with limited pick-up rear response. It is this tighter rear pattern that allows the most gain-before-feed back when used with in-house speaker systems.



Super Cardioid



"Club shaped" super directional characteristic



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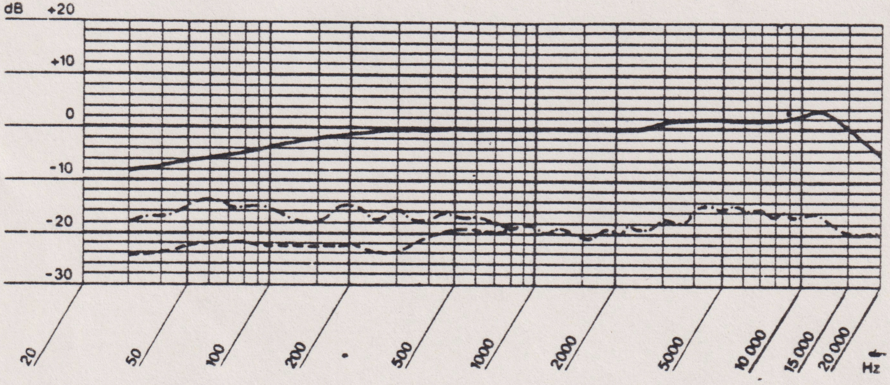
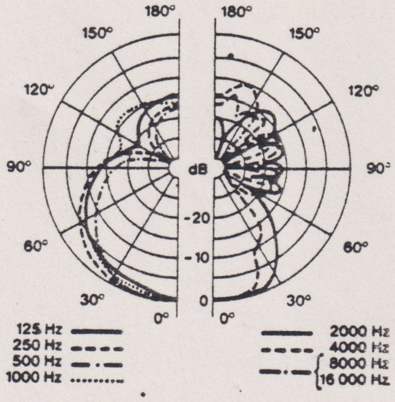
## Microphone Directional Characteristics. (Continued)

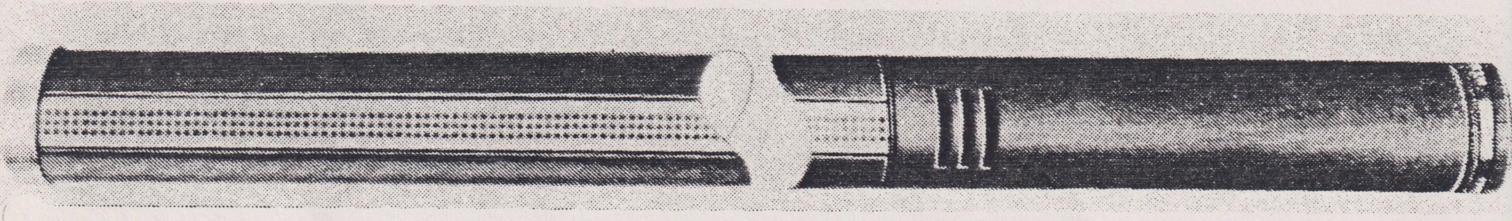
Both the cardioid and supercardioid allow greater working distances than the omnidirectional microphone, they also are used for boom operation and with their rear pattern limit reverberation.

### (iv) Highly Directional Microphones. (Shotgun)

The shotgun microphone is designed so that sound from the sides entering the sound pickup tube (figure 8) will cancel itself out automatically before it reaches the microphone at the end of the tube. This affect gives a highly directional forward direction pickup characteristic. Different length pickup tubes will affect how narrow the pickup pattern will be.







## Microphone Choice.

Omnidirectional when you require:-
- all round pickup pattern
- pickup of room acoustics
- extended low frequency response
- low handling noise
- low wind noise
- no up close bass boost

Unidirectional when you require:-
- selective pickup
- rejection of sounds behind the microphone
- rejection of room acoustics and leakage
- more gain-before-feedback
- up-close bass boost (proximity effect)

## Microphone Accessories.

(i) Power Supply Units (Phantom). Use to supply power to condensor microphones, some fitted with electronic roll-off filters and some with multiple microphone inputs/outputs.

(ii) Stands - because of handling problems with some of the microphones, shock mount stands stop noise being transmitted from desk through stand to microphone. Available in a number of different suspension and support systems.

(iii) Windscreens - or windshields can be fitted to most microphones and are used to cut down the effects of wind turbulence over the microphone in an outdoor situation or when indoors to reduce the effects of hard breathing sounds.

## Microphone Working Range.

When a performer stands beyond the working distance allowed by the microphone, the sound will become over reverberating, hollow and distant. The possibility of picking up excessive background noise is also increased. The recorded signal will be reduced and the signal-to-noise ratio will also be reduced, thus giving a poor playback signal.

## Microphone Impedance.

A microphone's impedance is its effective source resistance at 1000 Hz. Microphone impedances vary, those with an impedance from 150 to 600 chms are considered low impedance microphones, from 1000 to 4000 are medium, and above 25 kilohms are high impedance. The lower the impedance the longer the microphone cable you can use without hum pick-up or high frequency loss.

The load impedance of the microphone should be within the range of the input impedance of the device that the microphone is to be connected to (eg. mixer, tape recorder, V.C.R., etc.). To achieve this an impedance matching transformer is used for low to high impedance etc.

Low impedance balanced microphones can be connected with up to 200m of two conductor shielded cables without the addition of step-up transformers. For high impedance equipment it is essential to use a step up transformer at the microphone end of the cable.

High impedance microphone cable length should be kept to 2 m between microphone and equipment input.

## Balanced and Unbalanced Microphone Cables.

Figure 9 shows an unbalanced line where the signal is passed down the centre conductor (referred to as the "hot" lead) while the return path utilizes the cable shielding which is earthed at the equipment.
With this type of connection there is the possibility of interference in the form of hum pick-up in the cable on long distances, not so much a problem on short distances. Also try to keep cable runs as short as possible.

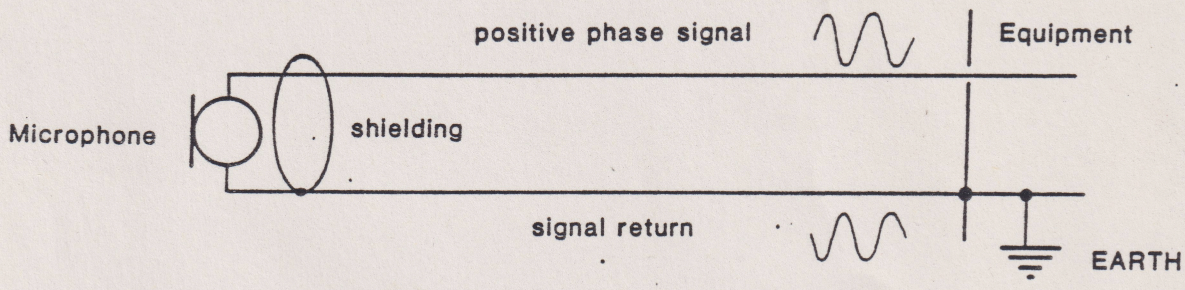

positive phase signal
Equipment
Microphone
shielding
signal return
EARTH

FIGURE 9

Figure 10 shows a balanced line where the signal is passed through one of the centre conductors while the other is used for the return path. The shielding is earthed at the equipment to give the best interference free signal.

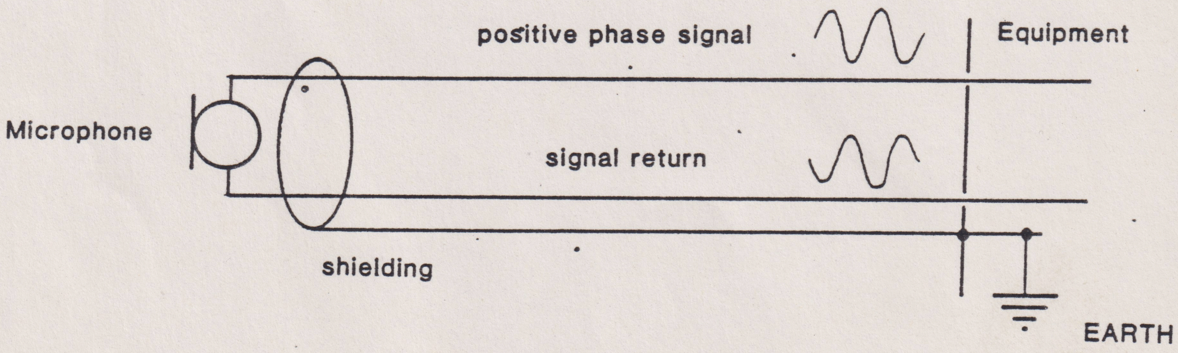

positive phase signal
Equipment
Microphone
signal return
shielding
EARTH

FIGURE 10

Note:- It is advisable to keep audio lead away from power leads to preclude the possibility of noise pick up.

## Microphone Standard DIN Connections.

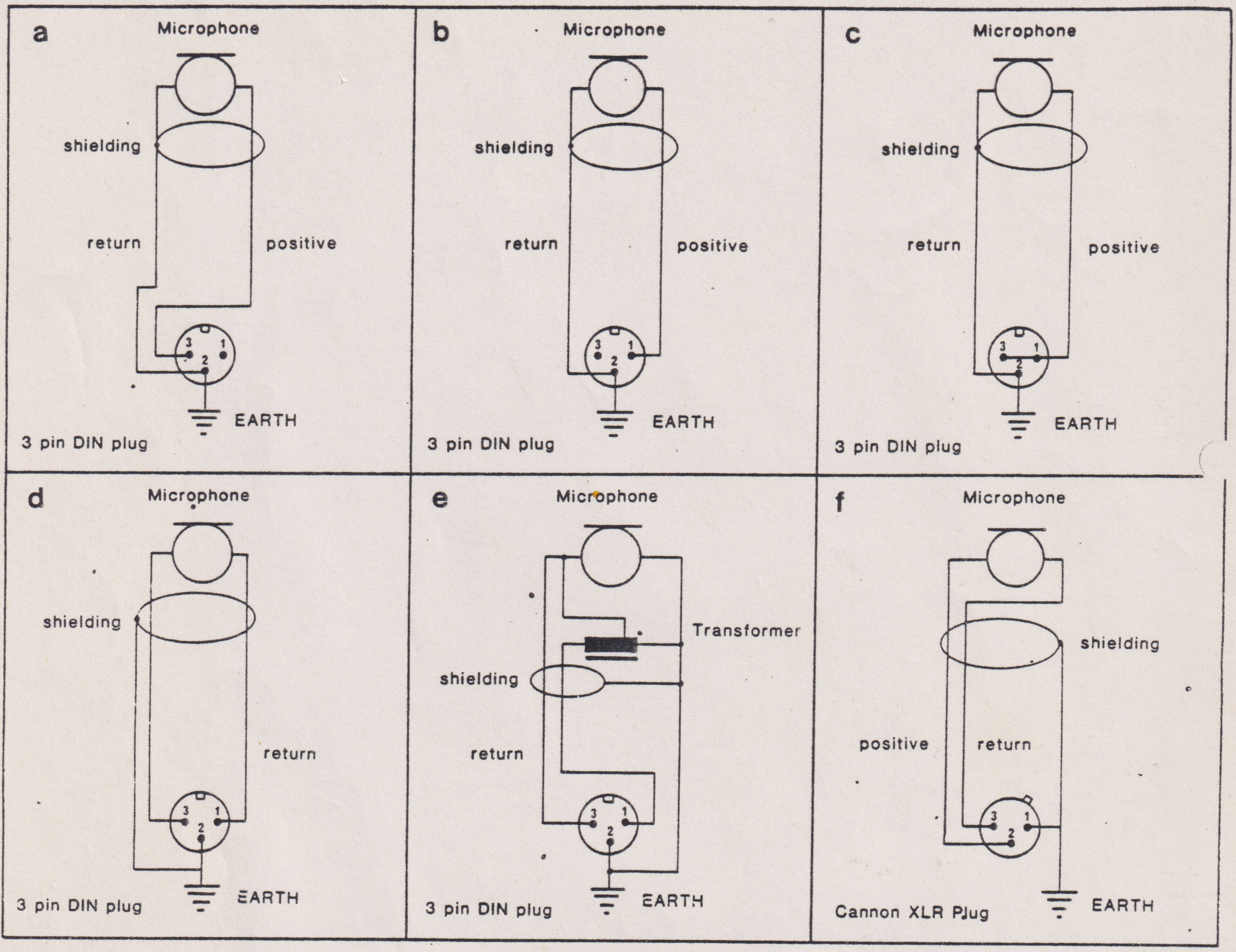

a
Microphone
shielding
return
positive
3
1
2
EARTH
3 pin DIN plug
b
Microphone
shielding
return
positive
3
1
2
EARTH
3 pin DIN plug
c
Microphone
shielding
return
positive
3
1
2
EARTH
3 pin DIN plug
d
Microphone
shielding
return
3
1
2
EARTH
3 pin DIN plug
e
Microphone
Transformer
shielding
return
3
1
2
EARTH
3 pin DIN plug
f
Microphone
shielding
positive
return
3
1
2
Cannon XLR Plug
EARTH

FIGURE 11

figure 11a - Low impedance unbalanced microphone

figure 11b - High impedance unbalanced microphone

figure 11c - Medium impedance microphone, the bridge between pins 1 and 3 makes it possible to use them in low impedance

figure 11d - Low impedance balanced microphones, pin 1, the postive signal phase wire.

figure 11e - Dual impedance microphones have inbuilt transformers and both high and low impedance outputs are available at the cable connector. Note, when used in the high impedance mode, cable length should be kept to 2m between microphone and equipment input.

...../11

3 GREEN
2 RED
1 GROUND

figure 11f - U.S. standard low impedance balanced microphone output fitted with Cannon XLR - type 3 pin male connector. Shield to pin 1 and connector, pin 2 positive signal phase and pin 3 signal return.

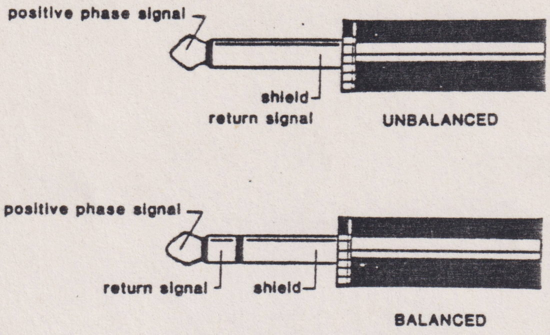

positive phase signal
shield
return signal
UNBALANCED
positive phase signal
return signal
shield
BALANCED

FIGURE 12

Figure 12 shows 6.3mm phone plug connections for balanced and unbalanced lines.

## Phasing of Multiple Microphone Installations.

When there is more than one microphone capable of picking up the sound, when connected through an audio mixer, then it is important that the microphones are wired properly at the plugs otherwise incorrect phasing of the signals will occur causing the signal from one microphone to reduce or completely cancel out the signal from the other microphone.

## AUDIO MIXERS

The function of the audio mixer is, as the name suggests, mixers audio signals together at a controlled amount so the sounds can be blended and the resulting signal switched to required outputs to be fed to amplifying or recording equipment.

In a simple mixer (figure 13) the amount of signal from inputs 1 and 2 can be individually controlled by a fader (volume) control. The resulting signal levels are feed into an electronic mixing circuit which will mix the two signals together without affecting the input circuits. The output of the mixing circuit is then feed to an amplifier whose output is controlled by a master fader. This master fader allows the inputs to be preset and the resultant mix evenly adjusted.

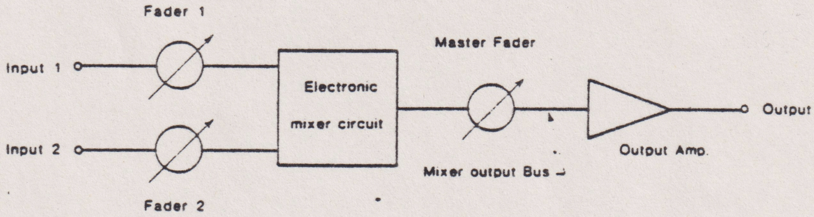

Fader 1
Input 1
Input 2
Fader 2
Electronic
mixer circuit
Master Fader
Mixer output Bus
Output Amp.
Output

figure 13

If two outputs are required, then the input required for each output is switched to the appropriate electronic mixer whose output is then feed through the output amplifier (figure 14).

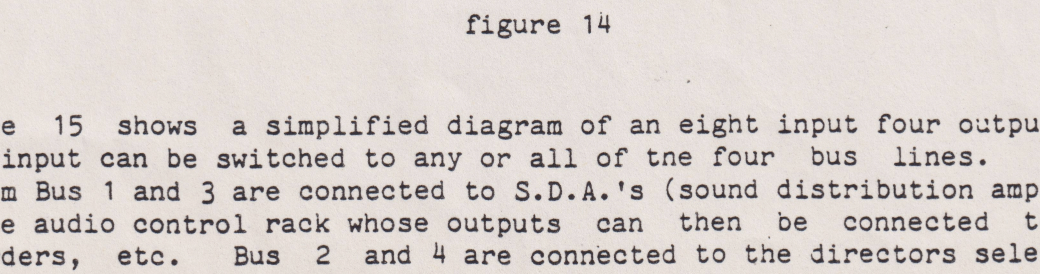

Faders
Input 1
Input 2
Input 3
Isolation Amp's
1
2
1
2
1
2
Output select
push buttons
Electronic
mixer circuit
Electronic
mixer circuit
Bus 1
Bus 2
Output Amp's
Master Fader
Output 1
Output 2

figure 14

Figure 15 shows a simplified diagram of an eight input four output mixer. Each input can be switched to any or all of tne four bus lines. In this system Bus 1 and 3 are connected to S.D.A.'s (sound distribution amplifiers) in the audio control rack whose outputs can then be connected to video recorders, etc. Bus 2 and 4 are connected to the directors selector for studio foldback (replay of audio to studio floor). All four buses are connected to the monitor section of the mixer, the output of which can be fed out to the control room amplifiers.

For simplification, the electronic mixing circuits and amplifiers have been left our of this diagram.

As outputs from various devices (microphones, video equipment, etc.) may vary, in voltage and impedance levels, so the input of the mixer should be capable of matching these devices. This is normally achieved by switching to microphone, phone and line.

The T.E.A.C. model 3 audio mixer has facilities to connect both low and high impedance microphones, low impedance (200 ohm) connected to inputs 1 to 6, high impedance (10k ohm) inputs 7 to 8. The nominal input level for microphone inputs is 3mV with a maximum level of 3V with the 40 dB attenuator inserted.

(dilute thin or weaken)

Attenuation (reduction) of the microphone signal level can be achieved by switching resistance into the microphone line to prevent the pre-amplifier stage from being overdriven by large signal levels which would cause distortion.

Phono inputs from a turn table can be connected to inputs 5 to 8 and they have an input impedance of greater than 45 K ohms.

Line inputs 1 to 8 have an input impedance of 20K ohm and a nominal input level of 0.3V and a maximum level of 15V.

## VU. METERS

THE VU (volume unit) meter is used to give a level indication so the signal does not exceed the recording equipment's signal-handling capacity which would result in distortion occuring. Figure 15 show a typical VU meter.

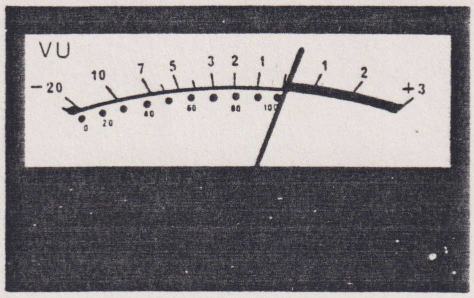


Figure 15.

VU meters have a dynamic characteristic that approximates the response of the human ear. The VU meter measures the audio output of each mixer board channel in volume units, a unit of measurement designed to read between the avergare and peak levels of complex audio wveforms such as music or speech. The reference level established for VU measurement is referred to as zero level (or zero VU).

In recording equipment the purpose of the VU meter is to set the signal level fed to the recording head to maximize the signal to noise ratio. If the level is too low then the signal to noise ratio decreases and the background noise becomes more apparent.

### Signal to Noise Ratio

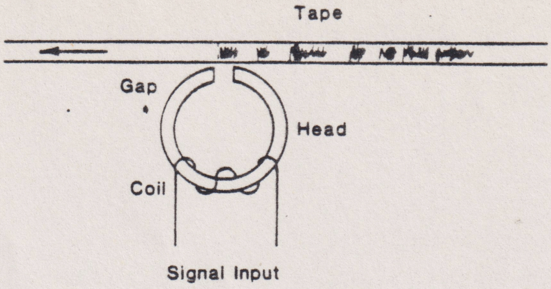
The signal to noise ratio (S/N) is a Decibel (dB) value defining how much higher the signal can be compared to the noise generated by the system without unacceptable distortion.

### Audio Tape Recording

An audio tape consists of a plastic backing material on which some form of magnetic sensitive particles have been deposited. The majority of tapes have ferrie oxide, though other types use chromium dioxide magnetic particals. This coating of magnetic particles is extremely thin (1 micron - one millionth of a meter).

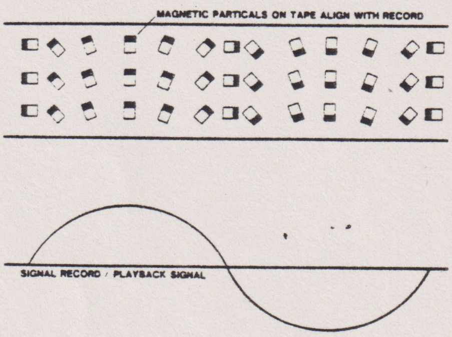
An important characteristic of a tape is the density and distribution of the magnetic particles, since it is these particles which will record the signal. If there is insufficient particles, the, when recording the complex audio signal parts would be missing, resulting in a poor recording and hence a poor sounding replay signal. It is therefore necessary to put as many and as evenly coated magnetic particles on the tape as possible.

Figure    shows an audio record head, the varying electrical signal is connected to the coil of the electromagnet. The varying signal causes a varying magnet flux to build up and decrease in the head gap and hence into the tape.



Figure

The varying signal aligns the magnetic particles to correspond to the record signal (figure ) and hence the electrical signal is now recorded magnetically on the tape.

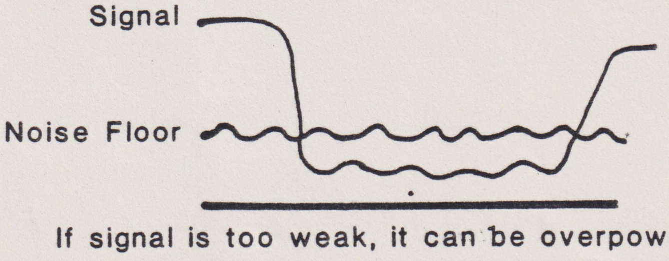

MAGNETIC PARTICALS ON TAPE ALIGN WITH RECORD
SIGNAL RECORD / PLAYBACK SIGNAL

Figure

On playback the magnetic information on the tape is replayed back over the same record head which now acts as the playback head. The magnetic signal on the tape induces small voltages, corresponding to the alignment of the magnetic particles, into the coil of the playback head. This signal is then amplified and then converted back into sound waves by the speaker.

Important factors effecting tape fidelity are the overall level of the signal produced on the tape to overcome tape noise; and the highest frequency that can be recorded on the tape. It is usually in the area of high frequency where the most noticeable performance fall off occurs with tape recorders.

The high frequences tend to be attenuated (weaken) because the time between fluctuations (cycles) are brief (figure ) because not enough tape has passed by the record head to allow sufficient space on the tape to record the cycle pattern.



**If signal is too weak, it can be overpowered by the noise floor of the tape.**

Figure

The following formula can be used to compare the amount of space available on a tape at a fixed frequency if the tape speed is changed.

$$L = S/F$$

where L = length of tape for recording
S = tape speed
F = frequency of the signal to be recorded.

Standard tape speeds are :-

4.75 cm/s
9.5 cm/s
19 cm/s
38 cm/s

If a 10kHz signal is to be recorded at each of the above tape speeds, then the length of tape that is available to record the signal on is:-

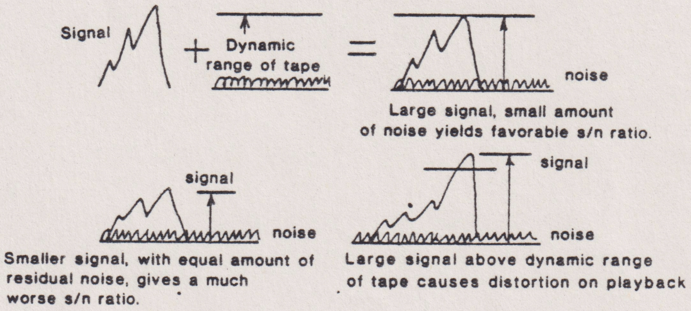
| Tape speed, | 4.75cm/s | 9.5cm/s | 19cm/s | 38cm/s |
|---|---|---|---|---|
| Tape length | 0.00475mm | 0.0095mm | 0.019mm | 0.038mm |

The higher the speed allows more detail (better fidelity for high frequency signals.

## Dynamic Range

The dynamic range of a program is the ratio between the strongest and softest sound in the program. For ochestral music this ratio is very large.

The dynamic range for the tape recorder is limited at the low end by the tape noise and at the upper end by distortion.



Large signal, small amount of noise yields favorable s/n ratio.

Smaller signal, with equal amount of residual noise, gives a much worse s/n ratio.

Large signal above dynamic range of tape causes distortion on playback

Figure

## Noise Reduction Techniques

Any piece of electronic equipment adds some degree of noise to the signal passing through it, but the major source of noise is from the magnetic recording tape. This is due to the very low level of signals from the tape and the inherent nolise property of the tape.

Along with the signal that is recorded there will be a continuous hissing sound, caused by the non alignment of some of the magnetic particles on the tape. The hiss is more or less at a constant level regardless of the recording level. If the recording signal matches the dynamic range of the tape, then if a sound occurs, the background hiss is drowned out, but for soft passages (record signal low), the tape hiss is quite audible.
There are a number of methods available to reduce the problem of noise, they are:-

(1) Equalization
(2) Dolby
(3) dbx.

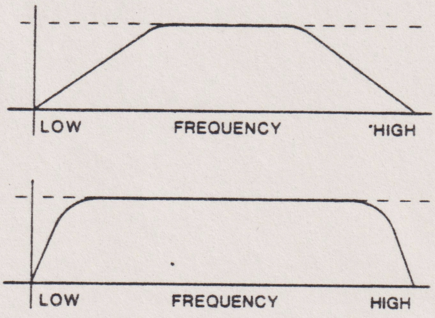
## (1) Equalization

The weakness of the induced voltage from the playback head increased the ever-present tape noise problem. This noise signal is typically rather small but because the desired signal is also very small then the noise signal becomes significant.

The induced voltage from the playback head is normally at its lowest for very low and very high frequencies, at such extreme frequencies the noise level tends to be more noticeable. Therefore, the signal to noise problem in the tape system is further aggravated by a combination of low recording signals (soft tones) at high frequencies.

To compensate for this low recorded level of the higher frequencies, some form of EQUALIZATION is usually applied to the signal before recording. Equalization is a deliberate alteration to the amplification of the signal, the lower and higher frequencies are given more amplification than the midrange frequencies. This allows the use of less equalization on playback, thus reducing the noise.

Care must be taken in the pre-recorded equalization process (if manually adjustable) or resulting tape/circuitry overload resulting in distortion of the signal can occur.



Figure

## (2) Dolby

There are three Dolby systems in general use developed by Dr. Ray Dolby, they are:-

(a) Dolby A
(b) Dolby B
(c) Dolby C

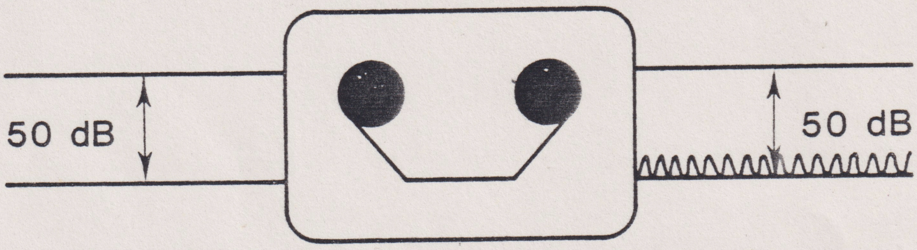
(a) Dolby A system is an expensive and complex professional N/R system and is mainly used in professional audio studios. This system splits the audio signal into four bands and then processes each band spearately, low frequencies (up to 60Hz) for hum and rumble problems, low midrange (60Hz to 3kHz) cross-talk and modulation, 3KHz to 9kHz and 9 kHz and above, for tapes hiss and modulation.

(b) Dolby B system used in cassette players for domestic and semi-professional use, works on the principle that if the low sound levels are raised in level before recording then on recording they will be above the noise level. On playback the level is turned down to what it was at the time of recording.

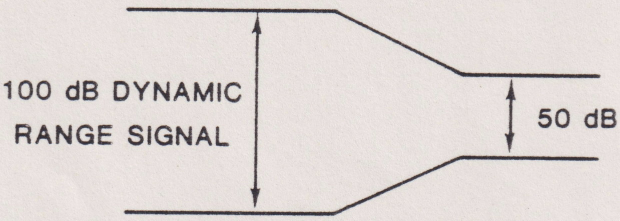
In the playback mode the high level signals (which wre unprocessed) mask the noise level on the tape while the medium - level higher frequencies are attenuated (reduced in level) to as close as possible to original levels. This attenuation of the signal will also reduce the noise signal. The low-level signal, because it ha maximum boost on record now requires maximum attenuation which wil reduce the noise to minimum value.

## (3) dbx System

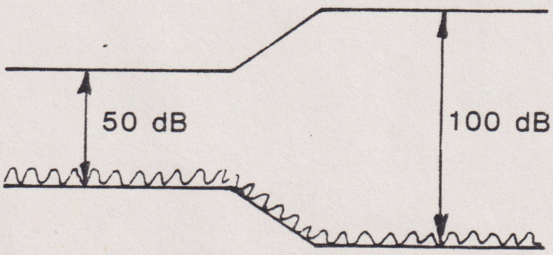
This is a straight compression - expansion type system with some pre-emphasis and de-emphasis. In record the signal is precisely compressed in a 2:1 ratio (figure ). This compression effectively doubles the dynamic range of the tape system, then on playback the dynamic range off the tape is expaned (doubled) to bring the sound back to the original levels.



SIGNAL PICKS UP NOISE WHEN RECORDING.



RECORD SIGNALS DYNAMIC RANGE
COMPRESSED BY 2:1 BEFORE RECORDING.



1:2 expansion on replay tape noise reduced

Compressed before
recording 2:1 100-50db
Expanded after
recording 2:1 50db-100db
So noise floor is ↓

AG:LAH

# TELEVISION EQUIPMENT NOTES.

## HIGH AND LOW LEVEL AUDIO INPUTS:

### Low Level Sources:

Un amplified signals that are of a very low level, e.g. Microphones, turntables.

### High Level Sources:

Amplified signals from equipment, e.g. Audio outputs from recorders.

## HOW TO IDENTIFY HIGH AND LOW LEVEL INPUTS AND OUTPUTS:

Low Level Inputs can be identified by Mic, phono.

High Level inputs/outputs can be identified -

Audio in/out
Line in/out

Note: Some equipment has multi level inputs, e.g. BVU Recorders use common input plugs with a High/Low level switch on the recorder.

## EFFECT OF WRONG CONNECTIONS:

Often it is not possible to connect High and Low levels together because of the intended non-compatibility of connecting plugs and sockets.

However if the two are connected together -

High to Low input - distortion due to overdriving, too much gain.

Low to High input - very low audio levels, not enough gain.

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## STUDENT TEST SHEET

## AUDIO MIXERS AND NOISE REDUCTION

1. Draw a simple diagram of a two input, one output mixer.

2. Mixers have two levels of input, explain.

3. What function does the mix attenuation switch perform?

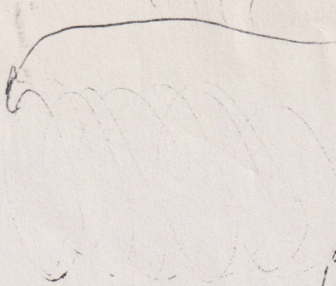
4. What is a satisfactory reading of a VU meter for correct recording.

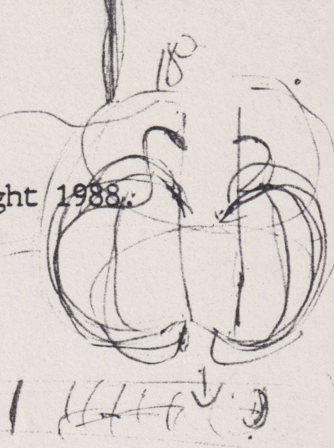
5. Explain Signal to Noise Ratio.

6. Explain the term "Dynamic Range".

7. Explain how the following Noise Reduction Techniques operate:-

   (a) Equalisation
   (b) Dolby
   (c) dBX





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## STUDENT TEST SHEET

## MICROPHONES

1. Name the three main types of microphone available.

2. Describe construction of the above three.

3. Which of the three types is most sensitive?

4. Which types of microphone require a battery or power supply?

5. Draw the response patterns for the following microphones:-

   (a) omnidirectional
   (b) birdirectional
   (c) directional (Cardiod)
   (d) shotgun.

6. What type of microphones would you select for the following situations:-

   (a) Public address on microphone stand; Shotgun
   (b) Background sound on portable video shoot; omni.
   (c) News gathering (outside);
   (d) Drama;
   (e) Interview (two way);
   (f) Long distance from mic.

7. Describe some difficulties in using microphones at a distance.

ohms
150—600 1,000—4,000

8. Give typical figures for impedances of low, medium and high impedance microphones.

↑25,000 ohms.

9. What advantages do low impedance microphones have?

10. Explain the difference between balanced and unbalanced lines.

11. Draw the connections of a balanced line connected to a Cannon XLR connector.

12. Explain what is meant by microphone phasing. One overrides the other.

correctly wired

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BENTLEIGH EAST, VIC. 3165

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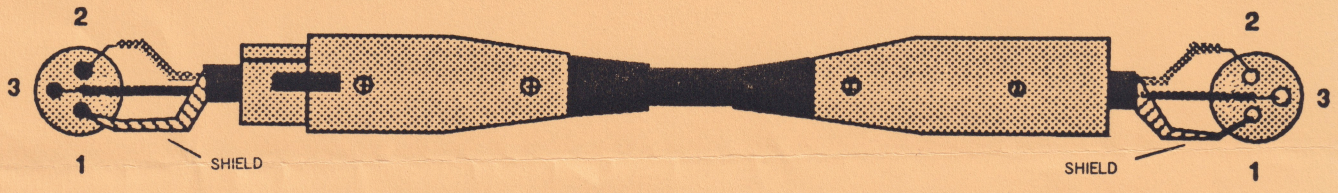
# HELPFUL HINT

## MICROPHONE CABLES

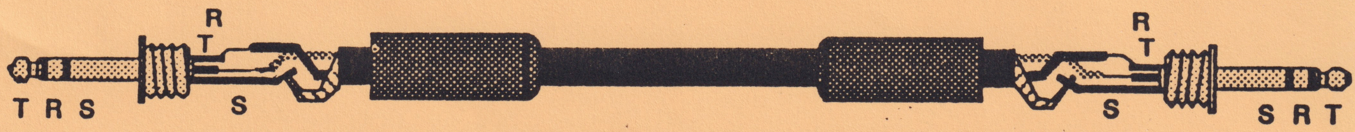
Balanced microphone cables have the advantage of canceling any noise which may be introduced into the signal over long cable runs.



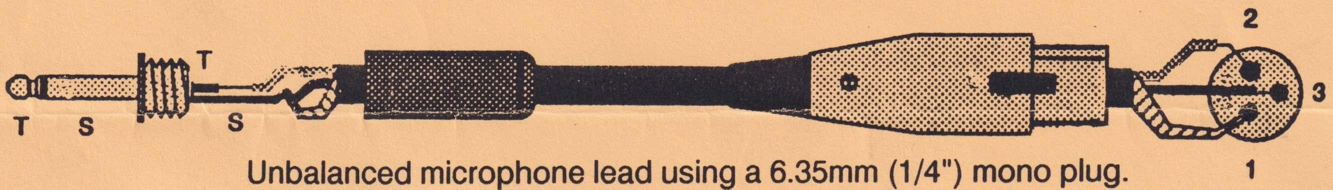
XLR 3 PIN FEMALE

XLR 3 PIN MALE

A standard balanced microphone lead is made up of a length of twin conductor shielded cable, with XLR style 3 pin female and male connectors at either end. The female (with button) plugs into the microphone while the male plugs into the mixer or amplifier.



Balanced microphone lead using 6.35mm (1/4") stereo plugs.



Unbalanced microphone lead using a 6.35mm (1/4") mono plug.

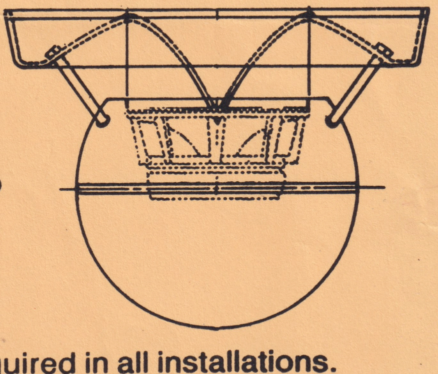
## IMPEDANCE MATCHING

In 70 and 100V applications, to achieve maximum amplifier efficiency, it is best to match the impedance of the speaker system to the output impedance of the amplifier.

The maximum permissible load for our SA series Amplifiers are tabled as follows:-

| | SA30 | SA60 | SA120 | SA120B | SA250B | SA500B |
|---|---|---|---|---|---|---|
| OUTPUT POWER | 30 watts | 60 watts | 120 watts | 120 watts | 250 watts | 500 watts |
| MAX LOAD (70V) | 163 Ohms | 82 Ohms | 41 Ohms | 41 Ohms | 20 Ohms | 10 Ohms |
| MAX LOAD (100V) | 333 Ohms | 170 Ohms | 80 Ohms | 80 Ohms | 40 Ohms | 20 ohms |

## WHERE DO YOU USE SOUNDSPHERES?



Soundsphere loudspeakers achieve outstanding projection of clear voice and high quality music in many challenging locations. These loudspeakers have been acclaimed for crisp voice announcements in shopping centres, racetracks, airports, school halls and churches.

Using the sphere/reflector combination, sound is reflected in a smooth circular pattern giving coverage over 360 degrees. Due to this extremely wide coverage, fewer speakers are required in all installations. Also, less amplifier power is needed, and the ease of installation can only result in cost savings to the installer.

Knowledge > KBase > Educational Materials > Connecting to a Shure Mixer

- Microphones
- Wireless
- Personal Monitors
- Mixers/DSP
- Personal Technology
- Discussion Systems
- Phono
- Accessories
- Product Literature
- Product News

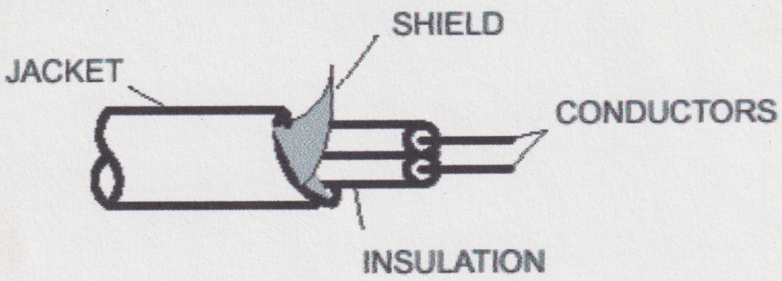
- KBASE
- Contractor's Corner
- Shure Notes
- Selection Guides & Comparison Charts
- User Guides
- Product Registration
- Service & Repair

# connecting your devices to a shure mixer

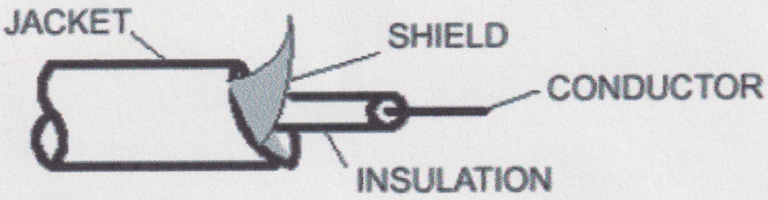
The variety of connectors on audio equipment sometimes leads to confusion in wiring. The diagrams below provide wiring recommendations for most common cables.

**NOTE:** XLR pin 1 is always connected to cable shield.

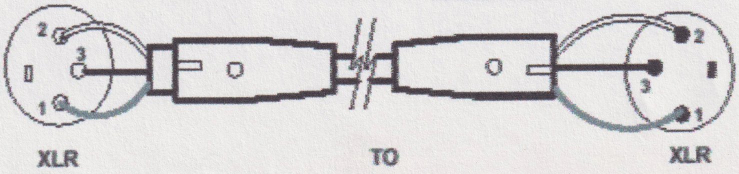
## BALANCED CABLE


SHIELD
JACKET
CONDUCTORS
INSULATION

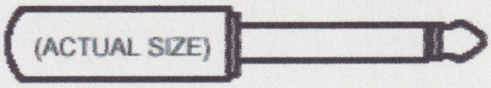
## UNBALANCED CABLE

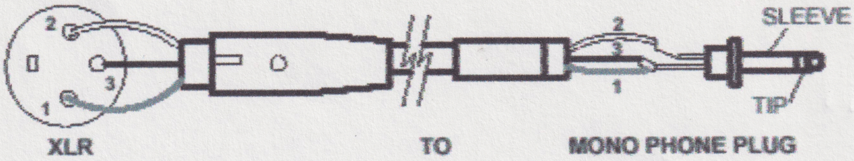

JACKET
SHIELD
CONDUCTOR
INSULATION

XLR CONNECTOR
(ACTUAL SIZE)

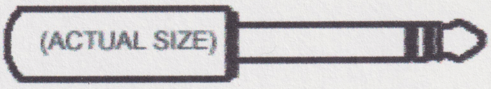

2
3
1
2
3
1
XLR
TO
XLR

**MONO PHONE PLUG**

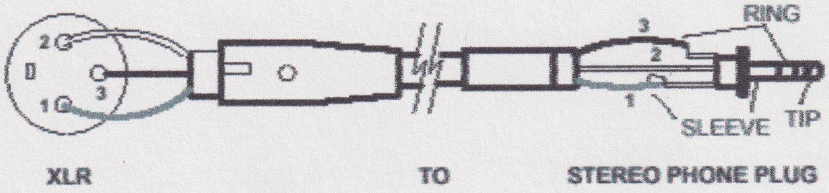

(ACTUAL SIZE)


2
3
1
SLEEVE
TIP
XLR
TO
MONO PHONE PLUG

**STEREO PHONE PLUG**

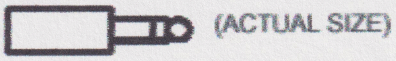

(ACTUAL SIZE)

**(USED AS BALANCED LINE CONNECTION)**


2
3
1
RING
SLEEVE
TIP
XLR
TO
STEREO PHONE PLUG

---

**3.5 MM (MINI) PHONE PLUG**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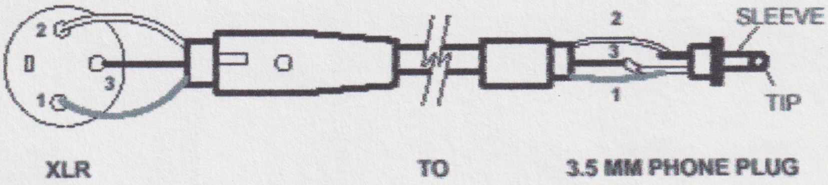
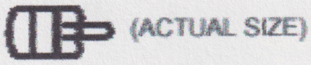
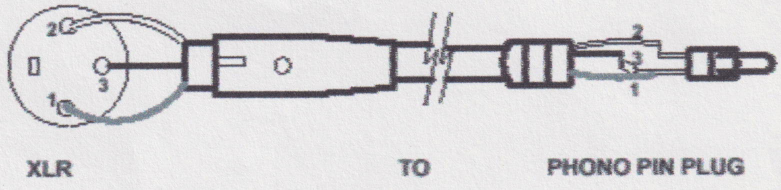

2
3
1
SLEEVE
TIP
XLR
TO
3.5 MM PHONE PLUG

---

**PHONO PIN (RCA) PLUG**


(ACTUAL SIZE)


2
3
1
XLR
TO
PHONO PIN PLUG

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